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| EXAMINER | |
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| LAI, ANDREW | |
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**Please find below and/or attached an Office communication concerning this application or proceeding.**

The time period for reply, if any, is set in the attached communication.

| Office Action Summary | Application No. | Applicant(s) | |
|---|---|---|---|
| | 10/776,894 | MINHAZUDDIN, MUNEYB | |
| | Examiner | Art Unit | |
| | Andrew Lai | 2616 | |

*-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --*

**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

**Status**

1)☒ Responsive to communication(s) filed on *10 February 2004*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

**Disposition of Claims**

4)☒ Claim(s) *1-41* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-41* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

**Application Papers**

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *10 February 2004* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

**Priority under 35 U.S.C. § 119**

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

**Attachment(s)**

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date *5/24/04,2/10/04*.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

## DETAILED ACTION

### *Claim Objections*

1. Claims 16/38 and 17/39 are objected to because of the following informalities: All these claims recite the limitation "*bandwidth utilization level*". This limitation appears to be a mistake which should be replaced instead by "available bandwidth level" and subsequent Office Action will be based thereon. Appropriate correction is required.

### *Claim Rejections - 35 USC § 103*

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims 1-9,12,14-15,17-18,20-22,23-31,34,36-37,39-40,42 are rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer et al (US 7,023,839, Shaffer hereinafter) in view of Graham et al (US 6,097,722, Graham hereinafter).

Shaffer discloses "system and method for dynamic codec alteration" (col. 1 lines 1-2) in "telecommunication systems" (col. 1 lines 16-17 and fig. 1) comprising the following features:

- **With respect to Independent claims 1 and 23**

**Regarding claims 1**, *a method for performing call admission control* (refer to fig. 1 and see "the H.323 gateway 106 generally provides ... and performs call setup and

clearing", col. 4 lines 41-44, see further, e.g. fig. 6 depicting "ARQ 602" for call Admission Request), *comprising: ...*

**Regarding claim 23**, *a call admission controller* (refer to fig. 1 "gatekeeper 108 and "BWAS [bandwidth allocation server] 109"), *comprising:*

*a processor* (fig. 3 "control processor 302") *operable to ...*

**Regarding claims 1/23**:

*(a) determining/determine* for the system *at least one of (i) a bandwidth utilization level* (refer to fig. 1 and see "the BWAS [bandwidth allocation server] 109 monitors system bandwidth usage"; *(ii) an available bandwidth level* ("the BWAS 109 calculates the remaining network bandwidth", col. 6 line 21); *and (iii) one or more Quality of Service or QoS metrics* ("the BWAS 109 saves the requested QoS levels for existing calls as well as the actual QoS level being provided", col. 9 lines 28-30);

*(b) comparing/compare the at least one of (i) a bandwidth utilization; (ii) an available bandwidth level; and (iii) one or more Quality of Service or QoS metrics to one or more selected thresholds* (generally see "BWAS 109 can measure and track the network traffic to make the determinations of the relevant thresholds being crossed", col. 5 lines 23-25, and particularly see "compares the system bandwidth usage against the threshold value X", col. 5 line 67 - col. 6 line 1, and "[check] if the threshold X were to be exceeded such that 1 Mbps network bandwidth is remaining", col. 6 lines 34-35; and further "if one or more new calls require a higher QoS, then the BWAS 109 determines whether lower QoS calls may be reset", col. 5 lines 30-33) *to determine whether a new live voice communication* ("performs call setup and clearing on both the

LAN side and switched circuit network (e.g., public switched telephone network or PSTN) side", col. 4 lines 43-45, noting "PSTN" deals with *live voice communications*) *may be set up with a first selected codec* ("the BWAS 109 monitors system bandwidth usage and directs each H.323 terminal to adopt a particular codec or coding algorithm according to bandwidth availability", col. 3 lines 5-8);

*(3) when a new live voice communication may not be set up with the first selected codec* (fig. 8 step 806 "BW [bandwidth] Avail ?" and associated "No" branch), *performing at least one of the following steps:*

*(i) selecting/select a second different codec from among a plurality of possible codecs for the new live coice communication, wherein the second codec has a lower bit rate than the first codec;*

*(ii) changing/change an existing live voice communication from the first codec to the second codec; and*

*(iii) redirecting/reditect the new live communication from the first path to a second different path, wherein the second path does not include the first link*

(fig. 8, continuing along above cited "No" branch, at step 812 where it is checked "if there exist connections whose QoS is presently more than needed or requested", col. 9 lines 44-46, and if "No", step 816 "make call with lower codec speed"; or if "Yes", step 814 "re-negotiate codec speed" for *existing communications* as further explained "If, however, the existing connections may be downgraded, the renegotiate lower codec speed process is undertaken in a step 814, ... and the call is made (step 808)", col. 9 lines 51-54).

Shaffer does not expressly disclose, regarding claims 1/23, that the step (a) determining/determine one of the bandwidth utilization level, available bandwidth level, and QoS metrics is with respect to *a* particular *path including a* particular *link.*

Graham discloses "bandwidth management processes and systems for asynchronous transfer mode networks using variable virtual paths" (col. 1 lines 1-4) wherein call "connection admission control" (col. 2 line 36) is implemented, comprising determining available bandwidth level with respect to *a particular path including a particular link* ("when a virtual channel connection is requested, it must be placed in a virtual path, so that the CAS software can determine if there is enough bandwidth Remaining in the virtual path to support the new virtual channel connection", col. 2 lines 36-40).

It would have been obvious to one of ordinary skill in the art at the time of the invention to modify the method/system of Shaffer by adding the path specific bandwidth determining mechanism of Graham to Shaffer in order to provide more efficient system "which allows for greater use of the available capacity of networks, and particularly, transmission facilities with a network" (Graham, col. 4 lines 24-26).

- **With respect to Dependent claims**

Shaffer discloses the following features:

**Regarding claims 2/24**, *(c) (i) is performed and further comprising:*

*receiving a request to place the live voice communication* (fig. 6 step 602 "ARQ" or "Admission Request");

*setting up the live voice communication with the second codec* (fig. 8 step 816 "make call with lower codec speed").

**Regarding claims 3/25**, *wherein each of a plurality of codecs has a corresponding bit rate and/or required bandwidth level* ("although the G.711 codec is the mandatory audio codec for an H.323 terminal, other audio codecs, such as G.728, G.729, G.723.1, G.722, MPEG1 audio, etc. may also be used", col. 3 lines 52-55, which codes are well known in the art to have different *bit rate* which in tern *require corresponding bandwidth level*) *and the selecting step comprises: comparing at least one of the available bandwidth level and the bandwidth utilization level with the plurality of bit rates and/or bandwidth level* ("if one or more new calls require a higher QoS (i.e., bandwidth), then the BWAS 109 determines whether lower QoS calls may be reset to still lower QoS codec", col. 5 lines 30-33); *and*

*selecting the highest quality codec having a corresponding bit rate and/or bandwidth level permitted by the at least one of the available bandwidth level and the bandwidth utilization level* ("the BWAS 109 may send an RAS command or H.245 signaling to the H.323 terminals to step down to the next fastest coding algorithm", col. 7 lines 45-47).

**Regarding claims 4/26**, *wherein the comparing comprises:*

*comparing at least one of (i) a bandwidth utilization level and (ii) an available bandwidth level with one or more selected thresholds; and comparing one or more Quality of Service or QoS metrics with one or more selected thresholds* (see discussion above regarding claims 1/23), *wherein the second codec has a bandwidth usage*

*characteristic sufficient to satisfy the comparing steps* ("the BWAS 109 may send an RAS command or H.245 signaling to the H.323 terminals to step down to the next fastest coding algorithm", col. 7 lines 45-47, noting that to be able to identify "the next fastest" codec, the codecs have to have *bandwidth usage characteristic sufficient* for *the comparing steps*).

**Regarding claims 5/27**, *wherein the comparing operation comprises:*

*estimating an impact on the one or more QoS metrics from placing the new live voice communication with the second codec* (refer to fig. 8 showing a series of steps when admitting a new call wherein step 802 shows "receive new QoS request level" determined primarily by the default codec to be used by the new call and then step 804 shows "compare requested QoS with available bandwidth", which step in combination with a later step 812 determining if existing "calls available" to use lowered codecs to meet "requested QoS with available bandwidth" will have to involve *estimating impact on QoS* due to the fact of *placing the new call with the second codec*, or "lower codec speed" shown in step 816 therein).

**Regarding claims 6/28**, *wherein, when there is no codec from among the plurality of codecs that satisfies the one or more thresholds, blocking the new live voice communication* (refer to fig. 6 and see "a calling H.323 terminal issues an Admission Request (ARQ) message to the gate keeper 108. In a step 604, the gatekeeper 108 accepts by issuing an Admission Confirm (ACF) message (it is noted that the gate keeper 108 could reject by responding with an Admission Reject (ARJ) message", col. 8 lines 18-22, noting that since Shaffer also disclosed lowering codecs as discussed

above, it would only be obvious, and in fact intuitive, to use ARJ for *blocking the new call when there is no codec from among the plurality of codecs* can be further lowered down to).

**Regarding claims 7/29**, *wherein substep (c)(ii) is performed* ("if the difference between the QoS levels meets a threshold, then the existing call(s) will have its (or their) codecs renegotiated to a lower level", col. 8 lines 65-57).

**Regarding claims 8/30**, *wherein, when the existing live voice communication was set up, the first and second codecs were identified as being acceptable to both endpoints.*

**Regarding claims 9/31**, *wherein substep (c)(ii) comprises:*

*renegotiating with destination the codec to be used in the live voice communication.*

("the BWAS ... monitors bandwidth usage, and if there is a disparity between the bandwidth allocated to new connections versus ongoing ones ... the BWAS sends a lower codec speed message to all active H.323 entities. This causes the H.323 entities to renegotiate their codecs. The original calling party then selects a lower Speed codec and sends a message to the called party to proceed with H.323 codec negotiation", col. 2 lines 9-17. It should be noted that the fact both parties were in "ongoing" call indicates that a first codec was accepted by both, and the "renegotiation" indicates that a second (and lower speed) codec is also accepted by both).

**Regarding claims 12/34**, *wherein in the determining operation the bandwidth utilization level is determined* (refer to figs. 1 and 3 and see "the BWAS 109, in

particular the bandwidth monitor 306, proceeds to monitor system bandwidth usage", col. 5 lines 62-64).

**Regarding claim 14/36**, *wherein the bandwidth utilization level is the end-to-end bandwidth* (since Shaffer discloses *end-to-end* codec negotiations, i.e. between calling and called parties, based on *bandwidth utilization level* monitoring, said *bandwidth utilization level* will also have to be *end-to-end*).

**Regarding claims 15/37**, *wherein in the determining step the available bandwidth level is determined* ("The BWAS 109 calculates the remaining network bandwidth", col. 6 lines 20-21).

**Regarding claims 17/39**, *wherein the available bandwidth level is the end-to-end bandwidth* ((since Shaffer discloses *end-to-end* codec negotiations, i.e. between calling and called parties, based on *available bandwidth level* monitoring, said *available bandwidth level* will also have to be *end-to-end*).

**Regarding claims 18/40**, *wherein one or more QoS metrics is determined* (fig. 8 step 800 "receive QoS levels").

**Regarding claim 20/42**, *wherein the available bandwidth level is the bandwidth allocable to live voice communications less the bandwidth utilization level* (this a obvious, and in fact intuitive, to one skilled in the art, because the term *available bandwidth* literally indicates remaining bandwidth that is not being used, in other words, *allocable bandwidth less the bandwidth utilization level*).

**Regarding claim 21**, *a computer readable medium* (fig. 3 "memory 308") *operable to perform the steps of claim 1* ("The control processor 302 is couple to a

memory 308 which is used to store bandwidth threshold information", col. 5 lines 8-10, noting that said "threshold information" is used to *perform the steps of claim 1* as already discussed above with respect to claim 1).

**Regarding claim 22**, *a logic circuit* (fig. 1 "gatekeeper 108" and "BWAS 109") *operable to perform the steps of claim 1* (see discussion above regarding claim 1).

Shaffer does not expressly disclose, regarding claims 18/40, the feature that QoS is determined *for the first path*, or in other words *path*-specifically determined, which feature is however disclosed by Graham (refer to fig. 1A and see "virtual paths are grouped or pooled together for Clients A and B by a number of factors, such as Quality of Service (QoS) and bandwidth", col. 5 lines 29-31).

4. Claims 10/32 and 11/33 are rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer (US 7,023,839, Shaffer hereinafter) in view of Graham et al (US 6,097,722, Graham hereinafter) as applied to claims 1 and 23 above, and further in view of Ho (US 6,452,922).

Shaffer in view of Graham discloses claimed limitations in paragraph 3 above.

Ho discloses that "an apparatus causes alternate connection of a telephone call directed an IP network" (Abstract lines 1-2) employing separate "IP interface" card and "PSTN interface" card (fig. 1 items 104 and 102).

Shaffer does not but Ho does disclose the following features:

**Regarding claims 10/32**, *wherein substep (c)(iii) is performed* [note: substep (c)(iii) recites *redirecting/redirect the new live voice communication from the first path to*

*a second different path wherein the second path does not include the first link*] (refer to fig. 1 and see "a network interface card 104 ... will cause a call directed to the card to be redirected to a different network 106 if the QoS for the call will be below the desired threshold", col. 2 lines 41-45).

**Regarding claims 11/33**, *wherein the first link corresponding to a first set of port numbers and the second link to a second set of port numbers, wherein the first and second sets of port numbers are non-overlapping, wherein packets addressed to one of the first set of port numbers are directed along the first link and packets addressed to one of the second set of port numbers are directed along the second link* (refer to fig. 1 wherein "call processor 100" comprises two separate, thus *non-overlapping*, ports communicating via two separate, thus *non-overlapping*, *links* with two separate, thus *non-overlapping*, network "interface" cards, i.e. "IP" and "PSTN". It is obvious as well as intuitive to one skilled in the art to appreciated for said separate or *non-overlapping ports* to have separate, thus *non-overlapping, first and second sets of non-overlapping port numbers*) *and wherein the redirection step comprises:*

*selecting/select for the packetized live voice communication a port address* (note that it is well known in the art that each port in a multi-ports communication unit, such as the "call processor 100" cited above, is identified by *a port address* or certain type of port ID) *from the first set of port numbers when a new live voice communication can be set up with the first selected codec and*

*selecting/select for the packetized live voice communication a port address from the second set of port numbers when a new live voice communication cannot be set up with the first selected codec.*

(still refer to fig. 1 and see "If the QoS that the IP network 1 108 has provided recently for test packets to be destination of the call to be routed is less than the desired QoS threshold, then the call is returned to the call processor 100 to be connected through another network 106 [noting that call to networks 108 and 106 uses above cited separate *ports, interfaces and links*]. Otherwise, the call is connected through the IP network 108).

It would have been obvious to one of ordinary skill in the art at the time of the invention to further modify the method/system of Shaffer by adding the call redirection mechanism of Ho to Shaffer in order to provide more robust system capable of "monitoring the quality of service (QoS) of the IP network and connecting a telephone call over an alternate network, on a call by call basis" (Ho, col. 1 lines 60-62) which would offer an important improvement to overcome the "disadvantage of VoIP Networks" having the "variability of the quality of the signal received at the destination as determined by changing network conditions" (Ho, col. 1 lines 42-44).

5. Claims 13/35 and 16/38 are rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer (US 7,023,839, Shaffer hereinafter) in view of Graham et al (US 6,097,722, Graham hereinafter) as applied to claims 1 and 23 above, and further in view of Lachman, III et al (US 2002/0166063, Lachman hereinafter)

Shaffer in view of Graham discloses claimed limitations in paragraph 3 above.

Lachman discloses "system and method for anti-network terrorism" (page 1 left column lines 1-2) using "a graph generated by Multi-Router Traffic Grapher (MRTG)" ([0140] lines 3-4).

Shaffer does not but Lachman does disclose the following features:

**Regarding claim 13/35**, *wherein the bandwidth utilization level is determined by polling a local edge router.*

**Regarding claims 16/38**, *wherein the available bandwidth level is determined by polling a local edge router.*

(refer to fig. 17, which "illustrates a man screen GUI for a central monitoring station", [0038] lines 1-2, and see "through block 1704, the MRTG can poll the Router's SNMP data and can chart the relative inbound/outbound bandwidth utilization", [0156] lines 4-6. It should be noted that knowing *bandwidth utilization* is equivalent to knowing *available bandwidth* because the latter is merely total bandwidth less the former).

It would have been obvious to one of ordinary skill in the art at the time of the invention to further modify Shaffer by adding the particular bandwidth monitoring method of Lachman to Shaffer in order to provide a better protected system "that can monitor incoming data packets fro a number of routers on a host network and that can detect a flood attack on an of the routers" (Lachman, [0014] lines 16-18).

6. Claims 19/41 are rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer (US 7,023,839, Shaffer hereinafter) in view of Graham et al (US 6,097,722,

Graham hereinafter) as applied to claims 18 and 40 above, and further in view of Melaku et al (US 2003/0074443, Melaku hereinafter).

Shaffer in view of Graham discloses claimed limitations in paragraph 3 above.

Melaku discloses "methods and systems for providing last mile quality of service for a network configuration with multiple access networks (Abstract lines 1-3).

Shaffer does not but Melaku does disclose the following features:

**Regarding claims 19/41**, *wherein the one or more QoS metrics is at lease one of packet delay, jitter, packet loss, the availability of Differential Services Code Point, and RSVP status* ("service level agreement is defined in terms of throughput, delay, jitter and packet loss", [0040] lines 3-4, "the LMQB service can map a bronze service onto differential services code point", [0069] lines 7-8), and "an end node or a source generates (1) an RCVP path message. This per-flow QoS signaling mechanism enables the end node to include parameters such as transmission rate, bandwidth, or other quality of service capabilities", [0072] 3-7).

It would have been obvious to one of ordinary skill in the art at the time of the invention to further modify the method/system of Shaffer by adding the various QoS metrics of Melaku to Shaffer in order to provide a enhanced QoS mechanism "for extending quality of service to the edge of a network" (Melaku, 0010] lines 5-6) to meet the "significant challenge" (Melaku, [0009] last line) of a "last mile" "bottleneck in delivery of data services to users" ([0009] lines 4-6) which "becomes even more severe when the data being delivered includes multimedia data because practical use of multimedia data requires a relatively large amount of bandwidth" ([0009] lines 8-11).

## *Conclusion*

7. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure.

US 6,581,031 discloses a speech compression system capable of encoding a speech signal using a plurality of codecs depending frame characteristics.

US 2003/0063569 provides a method for selecting a preferable codec mode for a connection between end UE's depending on network conditions.

US 2003/0123388 provides a method for call admission control depending on available bandwidth and adaptive using of multiple codecs per bandwidth characteristic on a per path/link basis.

US 2003/0074674 discloses method and system for dynamically adjusting video bit rates and thus adjusting coding/encoding rates.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Andrew Lai whose telephone number is 571-272-9741. The examiner can normally be reached on M-F 7:30-5:00 EST, Off alternative Fridays.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Kwang Yao can be reached on 571-272-3182. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

AL

KWANG BIN YAO
SUPERVISORY PATENT EXAMINER